## REMARKS

The present invention is a method of VoIP load management to assure voice quality in a packet switched network and a computer program embodied on a computer readable medium and executable by a computer for VoIP management to assure voice quality in a packet switched network. In accordance with an embodiment of the invention, a method of VoIP load management to assure voice quality in a packet switched network comprises determining a number of VoIP calls currently active in the packet switched network 220; determining a maximum number of VoIP calls the packet switched network can facilitate without loss of voice quality 230; allowing the admission of a new VoIP call when the addition of the new VoIP call would not exceed the maximum number of VoIP calls 240; and blocking the admission of a new VoIP call when the addition of the new VoIP call would exceed the maximum number of VoIP calls 250. The determination of the maximum number of VoIP calls the packet switched network can facilitate without loss of voice quality includes determining bandwidth for a plurality of communication links between a plurality of gateway pools 300, 310, 320, 330 and 340, determining the number of frames per IP packet used to transmit data in the packet switched network, and generating a capacity table indicating the maximum number of VoIP calls permitted for the plurality of communication links based on the bandwidth of each communication link and the frames per IP packet. See page 11 of the specification for an example of a capacity table. In accordance with an embodiment of the invention, a method of a VoIP load management to assure voice quality in a packet switched network comprises transmitting a ping request 520 to an originating

gateway 20 by a gatekeeper 50; transmitting a ping IP address to a destination gateway by an originating gateway 540; echoing a reply to the originating gateway by the destination gatekeepers 560; determining a roundtrip time for transmitting the ping request and echoing the reply 570; and allowing access of a new VoIP call to the packet switched network when the roundtrip time is less than a predetermined value 580.

Claims 1 and 7 stand rejected under 35 U.S.C. §102(e) as being anticipated by United States Patent 6,529,499 (Doshi et al) and further claims 2-12 stand rejected under 35 U.S.C. §103 as being unpatentable over Doshi in view of United States Patent 6,697,364 (Kekki et al). These grounds of rejection are reversed with respect to claims 2-6 and 8-12 with the subject matter of claims 2 and 8 having been incorporated into the subject matter of claims 1 and 7 except that the reference to "TRAU" frames has been inserted into newly submitted dependent claims 21 and 22.

Claims 1 and 7 recite, *inter alia*, determining the maximum number of VoIP calls the packet switched network can facilitate without loss of voice quality comprises determining bandwidth for a plurality of communication links between a plurality of gateway pools, determining the number of frames per IP packet used to transmit data in the packet switched network and generating a capacity table indicating the maximum number of VoIP calls permitted for the plurality of communication links based on the bandwidth of each communication link and the frames per IP packet. This subject matter is not disclosed by the combination of Doshi et al and Kekki et al.

It is submitted that Doshi et al does not disclose "determining bandwidth for a plurality of communication links between a plurality of gateway pools." While Doshi et al do disclose multiple gateways 215, such multiple gateways are not part of gateway pools as disclosed by Figs. 3 and 4 of the present application and Doshi et al do not disclose the determination of bandwidth for a plurality of communication links between a plurality of gateway pools. Moreover, there is no basis why Doshi would be modified by a person of ordinary skill in the art to provide gateway pools in place of the individual gateway units 215 and to further determine the bandwidth for a plurality of communication links between a plurality of gateway pools.

The Examiner relies on Kekki et al for showing the number of frames used to encode voice data for transmission directly relates to a capacity of calls that the system can accommodate. The Examiner asserts that Kekki et al teach the limitation of determining the number of frames per IP packet used to transmit data in the packet switched network and generating a capacity table indicating the maximum number of VoIP calls permitted for the plurality of communication links based on the bandwidth of each communication link and the frames per IP packet. In the first place, while Kekki et al do disclose a transcoder rate adapter unit (TRAU) as part of base station subsystem (BSS), such teaching does not suggest utilization in the claimed IP network. Kekki et al's teachings pertain to the GSM telephony system and do not have any specific disclosure regarding either determining the number of frames per IP packet used to transmit data in the packet switched network or generating the claimed capacity table.

Moreover, the Examiner's contention that "TRAU encoding is a fundamental quantity affecting how system capacity is used to transmit voice data" does not suggest the claimed generating a capacity table indicating the maximum number of VoIP calls permitted for the plurality of communication links based on the bandwidth of each communication link and the frames per IP packet. It is submitted that there is no teaching in either Doshi et al or Kekki et al suggesting the aforementioned limitation regarding the generating the capacity table as recited in claims 1 and 7.

Claims 3 and 9 further limit claims 1 and 7 in reciting accessing the capacity table whenever a new VoIP call requests entry to the packet switched network. As stated above, it is submitted that neither Doshi et al nor Kekki et al teaches the claimed capacity table and furthermore, it is submitted that there is no suggestion of accessing the capacity table whenever a new VoIP call requests entry to the packet switched network in either Doshi et al or Kekki et al.

Dependent claims 4 and 10 further limit claims 1 and 7 in reciting each gateway pool has in operation a plurality of communication devices connected to a gateway computer. It is submitted, as stated above, that there is no teaching of gateway pools in Doshi et al or Kekki et al. Moreover, there is no basis why a person of ordinary skill in the art would be led to modify the teachings of Doshi et al and Kekki et al to further connect a gateway computer to each gateway pool.

Dependent claims 5 and 11 respectively limit claims 4 and 10 in reciting that the plurality of gateway pools comprise at least one of the plurality of gateway pools having a gatekeeper which provides address translation and bandwidth management of the VoIP calls. As stated above with respect to the rejection of claims 1 and 7, there is no teaching in Doshi et al and Kekki et al of a plurality of

gateway pools and furthermore, it is submitted that there is no suggestion of gateway pools having a gatekeeper performing the function of translation and bandwidth management. There is no basis in the record why a person of ordinary skill in the art would be motivated to modify the teachings of Doshi et al and Kekki et al to arrive at the subject matter of claims 5 and 11.

Claims 6 and 12 further limit claims 5 and 11 in reciting that the gatekeeper manages access of VoIP calls to the packet switched network. Claims 6 and 12 are patentable for the same reasons set forth above with respect to claims 5 and 11.

Claims 13-20 stand rejected under 35 U.S.C. §103 as being unpatentable over United States Patent 6,515,964 (Cheung et al) in view of United States Patent 6,363,065 (Thornton et al). This ground of rejection is traversed for the following reasons.

Claims 13 and 17 respectively recite a method of VoIP load management to assure voice quality in a packet switched network and a computer program embodied on a computer readable medium and executable by a computer for VoIP load management to assure voice quality in a packet switched network comprising transmitting a ping request to an originating gateway by a gatekeeper; transmitting a ping address to a destination gateway by the originating gateway; echoing a reply to the originating gateway by the destination gatekeeper; determining a round trip for transmitting the ping request and echoing the reply; and allowing access of a new VoIP call to the packet switched network when the round trip is less than a predetermined value. This ground of rejection is traversed for the following reasons.

The Examiner erroneously concludes that Cheung et al suggest the utilization of a roundtrip delay. The Examiner refers to column 4, lines 5-63, for this teaching.

However, what is disclosed therein is "total delay from when one party utters a sound to when the other party hears that sound" as set forth in column 4, lines 18-20. What Cheung et al discloses is not roundtrip delay but merely the total time for transmission of the information in one direction. A person of ordinary skill in the art would not consider it obvious to utilize the claimed total roundtrip delay from the teaching of one-way transmission time referred to in Cheung et al. It is submitted that the references to "total delay" throughout column 4 and elsewhere in Cheung et al are in the context of the total delay for one-way transmission. If the Examiner persists in the stated grounds of rejection that Cheung et al teaches total roundtrip delay, it is requested that he specifically point out on the record where the basis is found in Cheung et al.

Moreover, the Examiner correctly recognizes that Cheung et al do not transmit a ping request or an echoing a reply. This follows since the claimed ping and echo pertain to the determination of two-way roundtrip delay which has no counterpart in Cheung et al. Since Cheung et al do not rely on roundtrip delay, the reliance upon Thornton et al for teaching "admission control by the gateways of the network may reference the latency of the network, measured by regularly sending a ping between peer gateways associated with the VoIP call (Col. 26, lines 10-14)" does not suggest to a person of ordinary skill in the art to utilize Ping messages in Cheung et al for measuring roundtrip transit time in the context of allowing access of a new VoIP call to the packet switched network when the roundtrip time is less than a predetermined value as recited in independent claims 13 and 17 is misplaced.

Moreover, the utilization of pinging in Thornton et al is for the purpose of switching an existing call from a private network to the PSTN which is different than

the claimed allowing access of a call to the packet switched network when the roundtrip time is less than a predetermined value.

Therefore, if the proposed combination of Cheung et al and Thornton et al were made, the subject matter of claims 13 and 17 would not be achieved since Cheung et al is based upon a different methodology of only measuring the one-way transmission time between a transmitter and a receiver and Cheung et al do not control admitting of calls to an IP network.

It is noted that the Examiner has not discussed the subject matter of dependent claims 14-16 and 18-20. These claims further recite further aspects of the claimed roundtrip time which is not suggested by the proposed combination of Cheung et al and Thornton et al since Thornton et al do not teach controlling access of a VoIP call to the packet switched network when the roundtrip time is less than a predetermined value and further, Cheung et al does not teach the usage of roundtrip time. Accordingly, the subject matter of the dependent claims 14-16 and 18-20 is not patentable.

Newly submitted claims 21 and 22 further limit claims 1 and 7 in reciting determining the number of frames is a number of TRAU frames which is patentable for the same reasons set forth above.

In view of the foregoing amendments and remarks, it is submitted that each of the claims in the application is in condition for allowance. Accordingly, early allowance thereof is respectfully requested.

To the extent necessary, Applicants petition for an extension of time under 37 C.F.R. §1.136. Please charge any shortage in fees due in connection with the

filing of this paper, including extension of time fees, to Deposit Account No. 01-2135 (0172.38632X00) and please credit any excess fees to such Deposit Account.

Respectfully submitted,

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Attachments
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